CLAIMS

WHAT IS CLAIMED IS:

1. An apparatus for supporting a plurality of data and voice services, the apparatus comprising:

a network interface configured to receive a call from a calling party device to a called party device;

signaling conversion logic configured to convert between Session Initiation Protocol (SIP) signaling and circuit-switched telephony signaling to support the call, wherein addressing information of the calling party device is preserved in the conversion, and the called party device includes one of a telephone station and a SIP client;

a voice port coupled to the signaling conversion logic and configured to communicate selectively with the telephone station; and

a data port configured to communicate selectively with the SIP client.

2. An apparatus according to claim 1, wherein the network interface receives a data stream, the apparatus further comprising:

a firewall logic configured to filter the data stream.

3. An apparatus according to claim 2, wherein the network interface communicates with one of a Digital Subscriber Line (DLS) network and a frame relay network.

4. An apparatus according to claim 2, wherein the data stream includes a plurality of packets, the apparatus further comprising:

Quality of Service (QoS) logic configured to classify the packets according to a predetermined QoS policy.

5. An apparatus according to claim 4, further comprising:

network management logic configured to detect and report fault within the apparatus, to monitor configuration information of the apparatus, to measure a network utilization parameter for billing and accounting, and to determine performance statistics.

6. An apparatus according to claim 1, further comprising:

another voice port communicating with a telephone switch configured to terminate the call to the called party station.

7. A method for supporting a plurality of data and voice services over a common customer premise equipment (CPE) device, the method comprising:

receiving a call from a calling party device to a called party device;

converting between Session Initiation Protocol (SIP) signaling and circuit-switched telephony signaling to support the call, wherein addressing information of the calling party device is preserved in the conversion, and the called party device includes one of a telephone station and a SIP client;

selectively communicating via a voice port interfacing the telephone station; and

selectively communicating via a data port interfacing the SIP client.

8. A method according to claim 7, wherein the network interface receives a data stream, the method further comprising:

filtering the data stream using a firewall.

9. A method according to claim 8, wherein the network interface communicates with one of a Digital Subscriber Line (DLS) network and a frame relay network.

10. A method according to claim 8, wherein the data stream includes a plurality of packets, the method further comprising:

classifying the packets according to a predetermined QoS policy.

11. A method according to claim 10, further comprising:

detecting and reporting fault within the CPE device;

monitoring configuration information of the CPE device;

measuring a network utilization parameter for billing and accounting; and

determining performance statistics.

12. A method according to claim 7, further comprising:

communicating with a telephone switch configured to terminate the call to the called party station.

13. A computer-readable medium carrying one or more sequences of one or more instructions for supporting a plurality of data and voice services over a common customer premise equipment (CPE) device, the one or more sequences of one or more instructions including instructions which, when executed by one or more processors, cause the one or more processors to perform the steps of:

receiving a call from a calling party device to a called party device;

converting between Session Initiation Protocol (SIP) signaling and circuit-switched telephony signaling to support the call, wherein addressing information of the calling party device is preserved in the conversion, and the called party device includes one of a telephone station and a SIP client;

selectively communicating via a voice port interfacing the telephone station; and

selectively communicating via a data port interfacing the SIP client.

14. A computer-readable medium according to claim 13, wherein the network interface receives a data stream, the computer-readable medium further including instructions for causing the one or more processors to perform the step of:

filtering the data stream using a firewall.

15. A computer-readable medium according to claim 14, wherein the network interface communicates with one of a Digital Subscriber Line (DLS) network and a frame relay network.

16. A computer-readable medium according to claim 14, wherein the data stream includes a plurality of packets, the computer-readable medium further including instructions for causing the one or more processors to perform the step of:

classifying the packets according to a predetermined QoS policy.

17. A computer-readable medium according to claim 16, further includes instructions for causing the one or more processors to perform the steps of:

detecting and reporting fault within the CPE device;

monitoring configuration information of the CPE device;

measuring a network utilization parameter for billing and accounting; and

determining performance statistics.

18. A computer-readable medium according to claim 13, further includes instructions for causing the one or more processors to perform the step of:

communicating with a telephone switch configured to terminate the call to the called party station.

19. A method for providing multiple communication services over a common interface device, the method comprising:

receiving telephony signaling pertaining to a call from a calling party according to a first signaling protocol compatible with a circuit-switched network;

generating a call setup message according to a second signaling protocol compatible with a data network;

determining whether the telephony signaling comprises address information pertaining to the calling party; and

inserting a header into the call setup message, the header containing a network address corresponding to the address information.

20. A method of claim 19, wherein the second signaling protocol is a Session Initiation Protocol (SIP), the method further comprising:

determining state of a screening indicator associated with the telephony signaling; and

responsive to said state, providing a value of a screening parameter in the call setup message.

21. A method of claim 19, wherein the second signaling protocol is a Session Initiation Protocol (SIP), the method further comprising:

determining a state of a presentation indicator associated with the telephony signaling; and

responsive to said state, providing a value of a privacy parameter in the call setup message.

22. A method of claim 19, wherein the second signaling protocol is a Session Initiation Protocol (SIP), the method further comprising:

determining state of a nature of address indicator in the telephony signaling;

responsive to the state of the nature of address indicator, formatting the address information pertaining to the calling party; and

providing the formatted address information in an address specification parameter in the call step up message.

23. A method of claim 22, further comprising:

identifying a trunk group associated with the telephony signaling;

determining a country code associated with the trunk group; and

including the country code in the formatted address information responsive to the state of the nature of address indicator.

24. A network device for providing multiple communication services, the device comprising:

means for receiving telephony signaling pertaining to a call from a calling party according to a first signaling protocol compatible with a circuit-switched network;

means for generating a call setup message according to a second signaling protocol compatible with a data network;

means for determining whether the telephony signaling comprises address information pertaining to the calling party; and

means for inserting a header into the call setup message, the header containing a network address corresponding to the address information.

25. A device of claim 24, wherein the second signaling protocol is a Session Initiation Protocol (SIP), the device further comprising:

means for determining state of a screening indicator associated with the telephony signaling; and

means for, responsive to said state, providing a value of a screening parameter in call setup message.

26. A device of claim 24, wherein the second signaling protocol is a Session Initiation Protocol (SIP), the device further comprising:

means for determining a state of a presentation indicator associated with the telephony signaling; and

means for providing a value of a privacy parameter in the call setup message responsive to said determined state.

27. A device of claim 24, wherein the second signaling protocol is a Session Initiation Protocol (SIP), the device further comprising:

means for determining state of a nature of address indicator in the telephony signaling;

means for formatting the address information pertaining to the calling party, responsive to the determined state of the nature of address indicator; and

means for providing the formatted address information in an address specification parameter in the call step up message.

28. A device of claim 27, further comprising:

means for identifying a trunk group associated with the telephony signaling;

means for determining a country code associated with the trunk group; and

means for including the country code in the formatted address information responsive to the state of the nature of address indicator.

29. A method for managing signaling in a communications system, the method comprising:

receiving a first signaling message compliant with a Session Initiation Protocol (SIP) and indicative of a call to a telephony system that uses a telephony signaling protocol that is not compliant with SIP;

creating a second signaling message according to the telephony signaling protocol; and

responsive to whether the first signaling message includes a remote party identification header, providing a calling party number element in the second signaling message, wherein the content of the calling party number element is derived from the content of the remote party identification header.

30. A method of claim 29, further comprising:

determining whether the first signaling message includes a screening parameter indicating that the calling party number is to be conveyed to the telephony system;

obtaining a configuration value indicating whether the calling party number is to be conveyed to the telephony system; and

selectively conveying the calling party number responsive to at least one of the configuration value and whether the first signaling message includes the screening parameter indicating that the calling party number is to be conveyed to the telephony system.

31. A method of claim 29, further comprising:

setting a privacy indicator in the second signaling message responsive to the value of a screening indicator in the first signaling protocol.

32. A method of claim 29, further comprising:

from the remote party identification header, identifying a trunk group in the telephony system to which the call is addressed;

determining a country code configured to be associated with the trunk group; and

providing the country code in the content of the calling party number.

33. A method of claim 32, further comprising:

determining a country designation in an address specification of the remote party identification header of the first signaling message;

comparing the country code to the country designation; and

selectively providing the country code responsive to whether the country code corresponds to the country designation.

34. A method of claim 29, wherein a SIP network server acts upon the first signaling message to communicate over a trunk group with a SIP client.

35. A method of claim 34, wherein privacy restrictions are conveyed from the trunk group to the SIP client.

36. A network device for supporting integrated voice and data services, comprising:

one or more voice ports configured to communicate with one or more analog devices;

one or more data ports configured to communicate with one or more Session Initiation Protocol (SIP) devices;

one or more network ports configured to communicate with a network;

firewall logic configured to filter traffic received from the one or more network ports; and

quality of service (QoS) logic configured to perform QoS processing on traffic received from the one or more voice ports, the one or more data ports, and the one or more network ports.

37. A network device of claim 36, further comprising:

a network address translator and a port address translator for translating network address and port information to communicate over the network.

38. A network device of claim 36, wherein the one or more voice ports include:

at least one Foreign Exchange Station (FXS)/Foreign Exchange Office (FXO) voice port, and

at least one of a T1 and fractional T1 trunk group.

39. A network device of claim 36, further comprising:

SIP/Time Division Multiplex (TDM) conversion logic connected to the at least one FXS/FXO voice port and the at least one of a T1 and fractional T1 trunk group and configured to translate signals between SIP and a TDM signaling protocol.

40. A network device of claim 36, wherein the analog devices include at least one of an analog telephone and a private branch exchange.

41. A network device of claim 36, wherein the one or more data ports include at least one Ethernet port.

42. A network device of claim 36, wherein the one or more network ports include at least one digital subscriber line (DSL) port and at least one frame relay port.

43. A network device of claim 36, wherein the filtering logic includes a firewall configured to filter the traffic received from the one or more network ports based on a set of rules.

44. A network device of claim 36, wherein the QoS processing includes:
classifying traffic received from the one or more voice ports, the one or more data ports, and the one or more network ports, and
scheduling the traffic received from the one or more voice ports, the one or more data ports, and the one or more network ports based on the classifying.

45. A network device of claim 36, further comprising:
a clock configured to generate a reference clock signal; and
an echo canceller configured to provide echo control and cancellation.

46. A network device of claim 36, further comprising:
network management logic configured to provide one or more of fault management, configuration management, accounting, performance management, and security for the network device.

47. A network device of claim 46 wherein, when providing fault management, the network management logic detects faults associated with the network device.

48. A network device of claim 46 wherein, when providing configuration management, the network management logic monitors configuration information associated with the network device.

49. A network device of claim 46 wherein, when providing accounting, the network management logic measures one or more network-utilization parameters.

50. A network device of claim 46 wherein, when providing performance management, the network management logic measures one or more parameters associated with network performance.

51. A network device of claim 46, wherein the network management logic is further configured to allow the network device to be remotely controlled and configured.

52. A network device for supporting integrated voice and data services, comprising:

at least one voice port configured to communicate with at least one analog telephone;

at least one voice trunk configured to communicate with a private branch exchange;

at least one data port configured to communicate with at least one Session Initiation Protocol (SIP) device;

at least one network port configured to communicate with a network; and

management logic configured to provide quality of service (QoS) management and security for the at least one voice port, the at least one voice trunk, the at least one data port, and the at least one network port.

53. A network device of claim 52, wherein the at least one voice port includes a Foreign Exchange Station (FXS)/Foreign Exchange Office (FXO) voice port.

54. A network device of claim 52, wherein the at least one voice trunk includes at least one of a T1 and fractional T1 voice trunk.

55. A network device of claim 52, further comprising:

a converter connected to the at least one voice port and the at least one voice trunk and configured to perform a bearer channel conversion and a signaling conversion.

56. A network device of claim 52, wherein the at least one data port includes an Ethernet port.

57. A network device of claim 52, wherein the at least one network port includes one or more of a digital subscriber line (DSL) port and a frame relay port.

58. A network device of claim 52, wherein the management logic includes a firewall configured to filter traffic received from the network.

59. A network device of claim 52, wherein the management logic includes a network address translator (NAT) and a port address translator (PAT).

60. A network device of claim 52, wherein the network device resides at a customer premises.